UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/964,825	09/27/2001	Yasuo Tezuka	100794-11728(FUJI 19.030)	2630

26304 7590 09/20/2007

KATTEN MUCHIN ROSENMAN LLP
575 MADISON AVENUE
NEW YORK, NY 10022-2585

EXAMINER	
NG, CHRISTINE Y	
ART UNIT	**PAPER NUMBER**
2616	

MAIL DATE	DELIVERY MODE
09/20/2007	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

TH

Office Action Summary	Application No.	Applicant(s)
	09/964,825	TEZUKA ET AL.
	Examiner	Art Unit
	Christine Ng	2616

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 02 July 2007.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-18 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-18 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 27 September 2001 is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claim 1-3 and 17 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 6,754,221 to Whitcher et al in view of U.S. Patent No. 6,738,351 to Qureshi et al.

Referring to claims 1 and 17, Whitcher et al disclose a gateway apparatus (Figure 1, gateway 18) which interconnects a first network (Figure 1, telecommunications network 12) and an IP network (Figure 1, data network 38). Data network 38 can be the Internet for delivering IP packets. Refer to Column 1, lines 26-36 and lines 51-57; and Column 4, lines 24-37. The apparatus comprises:

An encoding processing unit (Figure 2, compression module 108) receiving voice data from the first network and generating encoded voice data from the received voice data. Gateway 18 receives telecommunication information for the subscriber from telecommunication network 12 and compresses the telecommunication information according to the selected algorithm. Refer to Column 7, lines 49-55.

A packet processing unit (Figure 2, packetization module 110) creating IP packets of the encoded voice data from the encoding processing unit and transmitting the IP packets to the IP network, the packet processing unit receiving packets from a

second gateway apparatus (Figure 1; IAD 26, MTA 28 or WNIU 30) via the IP network. Gateway 18 then encapsulates the compressed telecommunication information, and communicates the data packets to the subscribers' customer premises equipment 14. Refer to Column 7, lines 55-59; and Column 11, lines 64-66.

A network-state estimation unit (Figure 2, memory 102) determining network-state information (bandwidth) of the IP network based on the received packets from the packet processing unit. Memory 102 stores a table (Figure 3) of customer premises information associating each customer premises equipment 14 with bandwidth and compression information. For communication with a particular customer premises equipment 14, a compression algorithm is chosen according to the available bandwidth. Refer to Column 12, line 39 to Column 14, line 4. Furthermore, as shown in Figure 5, the total bandwidth 302 and available bandwidth 308 determines what type of compression algorithm will be used for different subscribers 310-316. As subscribers 310-316 join the system, the bandwidth allocated for voice 306 increases and the available bandwidth 308 decreases. As the available bandwidth 308 decreases, gateway 18 selects the compression algorithm that uses less bandwidth. Therefore, the network-state information (bandwidth) is based on packets received from customers. If there are more customers sending packets, the available bandwidth decreases which affects the chosen compression algorithm. Refer to Column 15, line 15 to Column 16, line 19.

A determination unit (Figure 2, management module 100) controlling, before the transmission of the IP packets, at least the encoding of the voice data by the encoding

processing unit based on the network-state information (bandwidth) determined by the network-state estimation unit. Management module 100 manages the operation of gateway 18 using bandwidth, compression, and subscriber information stored in memory 102. For communication with a particular customer premises equipment 14, management module 100 determines a compression algorithm according to the available bandwidth. Refer to Column 9, lines 17-23 and lines 45-58; and Column 11, lines 1-20.

Wherein the IP packets to be transmitted to the IP network are processed according to network-state information indicating only the state of the IP network, independently of the network state of other networks. As shown in Figure 5, as subscribers 310-316 join the system, the bandwidth allocated for voice 306 increases and the available bandwidth 308 decreases. As the available bandwidth 308 decreases, gateway 18 selects the compression algorithm that uses less bandwidth. Therefore, the network-state information (bandwidth) is based on packets received from customers. If there are more customers sending packets, the available bandwidth decreases which affects the chosen compression algorithm. Only the state of the IP network (the number of users connected to the IP data network 38 and the bandwidth they are using) is used in determining the encoding of the packets. Refer to Column 15, line 15 to Column 16, line 19.

Whitcher et al do not disclose that the packet processing unit receives real-time transport control protocol (RTCP) packets.

Qureshi et al disclose in Figure 2 a similar system of controlling compression based on network conditions. NCM 38 determines the congestion of the system. When congestion decreases in the system, the voice compression ratio is decreased and when congestion increases in the system, the voice compression ratio is increased. Refer to Column 13, line 52 to Column 14, line 62. Congestion control mechanisms can be based on RTP measurements in accordance with the RTP/RTCP protocol. The RTP protocol allows for accurate packet loss and delay measurements. Refer to Column 15, liens 26-39. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include that the packet processing unit receives real-time transport control protocol (RTCP) packets. One would have been motivated to do so since RTCP packets allow the system to more accurately measure packet loss and delay, thereby better determining the network state.

Referring to claim 2, Whitcher et al disclose that the determination unit (Figure 2, management module 100) determines a type of the encoding (Figure 3, available voice compression algorithms 216) that is performed by the encoding processing unit, based on the network-state information (bandwidth) of the IP network. Based on the bandwidth between the gateway 18 and customer premises equipment 14, the management module 100 chooses a particular encoding method. Refer to Column 9, lines 17-23 and lines 45-58; Column 11, lines 1-20; and Column 12, line 39 to Column 14, line 4.

Referring to claim 3, Whitcher et al discloses wherein the determination unit (Figure 2, management module 100) determines an option of non-voiced data

compression or non-compression that is performed by the encoding processing unit, based on the network-state information of the IP network. Telecommunication information from telecommunication network 12 may include voice, data, image, video, or any other type of information. Refer to Column 3, lines 8-11. Also, packetization modules 110 may receive either compressed telecommunication information from compression modules 108 or uncompressed telecommunication information from TIMs 104 or echo cancellation modules 106. Refer to Column 11, lines 40-44.

3. Claims 4, 5 and 10 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 6,754,221 to Whitcher et al in view of U.S. Patent No. 6,738,351 to Qureshi et al, and in further view of U.S. Patent No. 6,760,309 to Rochberger et al.

Referring to claim 4, Whitcher et al disclose in Figure 2 that the determination unit (management module 100) controls the packetizing of the packet processing unit (packetization module 110). Refer to the rejection of claim 1.

Whitcher et al do not disclose wherein the determination unit determines a packet discarding priority level of the packet processing unit, based on the network-state information of the IP network.

Rochberger et al discloses in Figure 5, packets are assigned priority levels depending on their TTL values, with packets having a lower TTL being assigned lower priorities since they cannot be used at the destination. In Figure 8, steps 156 and 158, packets that arrive with TTL field values smaller than a threshold are discarded since they are stale to an extent that they cannot be used at the destination. Refer to Column 11, lines 44-65 and Column 14, lines 33-58. Therefore, it would have been obvious to

one of ordinary skill in the art at the time the invention was made to include that the determination unit determines a packet discarding priority level of the packet processing unit, based on the network-state information of the IP network. One would be motivated to do so in order to assign lower priorities to packets with smaller TTL values, so that the system can discard older packets to prevent network congestion and overflow.

Referring to claim 5, Whitcher et al disclose in Figure 2 that the determination unit (management module 100) controls the packetizing of the packet processing unit (packetization module 110). Refer to the rejection of claim 1.

Whitcher et al do not disclose wherein the determination unit determines a packet transmission priority level of the packet processing unit, based on the network-state information of the IP network.

Rochberger et al discloses a determination unit (Figure 6, processor 114) which determines a packet transmission priority level of the packet processing unit, based on the network-state information of the second network. Delay sensitive queues 108 and non-delay sensitive queues 106 are assigned priorities and are transmitted by the processor 114 according to priorities. The delay sensitive queues 108 are further broken down into different priority levels P1-P4, of which are transmitted according to Figure 8. Refer to Column 12, lines 29-60 and Column 14, lines 33-58. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the determination unit determines a packet transmission priority level of the packet processing unit, based on the network-state information of

the IP network. One would be motivated to do so in order to allow higher priority packets to be transmitted before lower priority packets.

Referring to claim 10, Whitcher et al do not disclose wherein the network-state estimation unit reads a TTL value from a packet that is received from the second gateway apparatus via the IP network at a start of communication, the network-state estimation unit sending the TTL value to the determination unit.

Rochberger et al discloses that each packet contains a TTL field that conveys the time left before the packet is no longer of any use in the network. Each network entity that receives the packet with a TTL field subtracts from it the time the packet spent in that entity. Thus, the TTL field decreases as it hops from network entity to entity in the network. As shown in Figure 8, steps 156 and 158, a receiving network element reads the TTL value from a received packet and uses it to determine the priority of the packet to place it in a corresponding queue or discard it if its TTL value is blow a threshold. Refer to Column 14, lines 33-58. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the network-state estimation unit reads a TTL value from a packet that is received from the second gateway apparatus via the IP network at a start of communication, the network-state estimation unit sending the TTL value to the determination unit. One would be motivated to do so in order to utilize a TTL value of a packet to determine how long a packet has been in the network, thereby discarding all older packets to prevent network congestion and overflow.

4. Claims 6 and 8 are rejected under 35 U.S.C. 103(a) as being unpatentable over

U.S. Patent No. 6,754,221 to Whitcher in view of U.S. Patent No. 6,738,351 to Qureshi et al, and in further view of U.S. Patent No. 6,868,094 to Bordonaro et al.

Referring to claim 6, Whitcher et al do not disclose wherein the network-state estimation unit determines a packet loss ratio based on the IP packets that are received from the second gateway apparatus via the IP network, and sends the packet loss ratio to the determination unit.

Bordonaro et al discloses in Figures 1A and 1B a network-state estimation unit (sender 18a/voice gateway 18a) that determines a packet loss ratio based on packets (probe packets Pa' and Pb') that are received from a second gateway apparatus (respsonder 18b/voice gateway 14b) via the second network and sends the packet loss ratio to the determination unit (sender 18a/voice gateway 18a). In Figure 4, sender 18a sends (step 100) probe packets containing a send sequence number field to responder 18a who modifies (step 102) the receive sequence number field and sends the probe packet back to sender 18a. Sender 18a can then use (step 104) the probe packet to determine packet loss. Refer to Column 3, lines 23-33; Column 7, lines 47-54; Column 9, lines 7-46; and Column 11, lines 28-55. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the network-state estimation unit determines a packet loss ratio based on the IP packets that are received from the second gateway apparatus via the IP network, and sends the packet loss ratio to the determination unit. One would be motivated to do so in order for the service provider to measure data packet loss and for the users to monitor data

packet loss to ensure a requested quality of service level. Refer to Column 1, lines 26-29.

Referring to claim 8, Whitcher et do not disclose wherein the network-state estimation unit determines a packet arrival time jitter based on packets that are received from the second gateway apparatus via the IP network, and sends the packet arrival time jitter to the determination unit

Bordonaro et al discloses in Figures 1A and 1B a network-state estimation unit (sender 18a/voice gateway 18a) that determines a packet arrival time jitter based on packets (probe packets Pa' and Pb') that are received from a second gateway apparatus (respsonder 18b/voice gateway 14b) via the second network and sends the packet arrival time jitter to the determination unit (sender 18a/voice gateway 18a). In Figure 4, sender 18a sends (step 100) probe packets containing a send time field to responder 18a who modifies (step 102) the receive time field and sends the probe packet back to sender 18a. Sender 18a can then use (step 104) the probe packet to determine packet jitter. Refer to Column 3, lines 34-44; Column 7, lines 37-46; Column 8, line 65 to Column 9, line 6; and Column 11, lines 28-55. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the network-state estimation unit determines a packet arrival time jitter based on packets that are received from the second gateway apparatus via the IP network, and sends the packet arrival time jitter to the determination unit. One would be motivated to do so in order for the service provider to measure data packet jitter and for

the users to monitor data packet jitter to ensure a requested quality of service level. Refer to Column 1, lines 26-29.

5. Claims 7 and 9 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 6,754,221 to Whitcher in view of U.S. Patent No. 6,738,351 to Qureshi et al in view of U.S. Patent No. 6,868,094 to Bordonaro et al, and in further view of U.S. Patent No. 6,816,464 to Scott et al.

Whitcher et al discloses that a set of predetermined control parameters levels being inclusive of at least one of a set of packet discarding priority levels (none), a set of packet transmission priority levels (none), and set of encoding type levels (claim 2). Refer to the rejection of claim 2.

Whitcher et al do not disclose that the determination unit stores at least one reference value of the packet loss ratio (claim 7) / the packet arrival time jitter (claim 9), and determines a specific one of a set of predetermined control parameter levels based on the result of comparison of said at least one reference value and the packet loss ratio (claim 7) / packet arrival time jitter (claim 9) received from the network-state information.

Scott et al disclose a network-state storage unit (Figure 3, database 310) that stores results of route tests, route checking parameters, and route information of various candidate routes with respect to a particular destination (Figure 2, gateway 204, 206 or 208). The network-state information includes average delay, average jitter and packet loss. Refer to Column 7, lines 11-40. The database 310 stores a reference value of a packet loss ratio and a packet arrival time jitter and uses the reference values

to compare test results (table 1 and table 2) of candidate routes. The reference value of the packet loss ratio and packet arrival jitter is obtained and stored from previous tests. Refer to Column 7, lines 41-65. When a call to a particular gateway is made, the database 310 determines a specific one of a set of predetermined control parameters (scoring the candidate routes, prioritizing the candidates routes, and choosing the best route) based on the comparison of the network-state information with the reference values. Refer to Column 8, line 33 to Column 9, line 50. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include that the determination unit stores at least one reference value of the packet loss ratio (claim 7) / the packet arrival time jitter (claim 9), and determines a specific one of the set of predetermined control parameter levels based on the result of comparison of said at least one reference value and the packet loss ratio (claim 7) / packet arrival time jitter (claim 9) received from the network-state information. One would be motivated to do so to compare network conditions of candidate routes with reference value in order to provide a basis for which to prioritize the routes and choose the best route for packet transmission. Network conditions also affect which encoding algorithm should be used to encode the packets. Compression algorithms providing higher quality require higher bandwidth utilization. Compression algorithms providing lower quality saves bandwidth but can affect the transmission quality.

6. Claim 11 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 6,754,221 to Whitcher et al in view of U.S. Patent No. 6,738,351 to Qureshi

et al in view of U.S. Patent No. 6,760,309 to Rochberger et al, and in further view of U.S. Patent No. 6,816,464 to Scott et al.

Refer to the rejection of claims 7 and 9 and claim 10. Furthermore, Whitcher et al and Scott et al do not disclose that the reference value is a TTL value.

Rochberger et al discloses in Figure 5, packets are assigned priority levels depending on their TTL values, with packets having a lower TTL being assigned lower priorities since they cannot be used at the destination. In Figure 8, steps 156 and 158, packets that arrive with TTL field values smaller than a threshold are discarded since they are stale to an extent that they cannot be used at the destination. Refer to Column 11, lines 44-65 and Column 14, lines 33-58. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include that the reference value is a TTL value. One would be motivated to do so since a packet's TTL value determines how long a packet has been in the network, thereby allowing older packets to be discarded to prevent network congestion and overflow.

7. Claims 12-15 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 6,754,221 to Whitcher in view of U.S. Patent No. 6,738,351 to Qureshi et al, and in further view of U.S. Patent No. 6,816,464 to Scott et al.

Referring to claim 12, Whitcher et al do not disclose a network-state storage unit storing the network-state information with respect to each of a plurality of destination stations in the IP network, wherein the determination unit stores a reference value of one of a packet loss ratio and a packet arrival time jitter, and, when a call connection between the gateway apparatus and one of the plurality of destination stations is

established, the determination unit determines a specific one of a set of predetermined control parameter levels based on the result of comparison of the reference value and the network-state information of said one of the plurality of destination stations read from the network-state storage unit.

Scott et al disclose a network-state storage unit (Figure 3, database 310) that stores results of route tests, route checking parameters, and route information of various candidate routes with respect to a particular destination (Figure 2, gateway 204, 206 or 208). The network-state information includes average delay, average jitter and packet loss. Refer to Column 7, lines 11-40. The database 310 stores a reference value of a packet loss ratio and a packet arrival time jitter and uses the reference values to compare test results (table 1 and table 2) of candidate routes. The reference value of the packet loss ratio and packet arrival jitter is obtained and stored from pervious tests. Refer to Column 7, lines 41-65. When a call to a particular gateway is made, the database 310 determines a specific one of a set of predetermined control parameters (scoring the candidate routes, prioritizing the candidates routes, and choosing the best route) based on the comparison of the network-state information with the reference values. Refer to Column 8, line 33 to Column 9, line 50. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include a network-state storage unit storing the network-state information with respect to each of a plurality of destination stations in the IP network, wherein the determination unit stores a reference value of one of a packet loss ratio and a packet arrival time jitter, and, when a call connection between the gateway apparatus and one of the plurality of destination

stations is established, the determination unit determines a specific one of a set of predetermined control parameter levels based on the result of comparison of the reference value and the network-state information of said one of the plurality of destination stations read from the network-state storage unit. One would be motivated to do so to compare network conditions of candidate routes with reference values in order to provide a basis for which to prioritize the routes and choose the best route for packet transmission.

Referring to claim 13, Whitcher et al does not disclose wherein the network-state estimation unit transmits test voice data to the second gateway apparatus via the IP network, receives test packets from the second gateway apparatus via the IP network, and determines the network-state information, including an estimated network delay and an estimated voice data quality level, based on the result of comparison of the test voice data and the test packets.

Scott et al disclose in Figure 5 a method of testing the network quality. The network-state estimation unit (routing manager 306) in the source gateway sends (step 508) test packets in the form of quality measurement packets to the destination gateway. The destination gateway receives (step 510) the quality measurement packets and returns the packet back to the originating gateway as soon as possible. The returned tests packet includes information about the packet that was received by the destination gateway. The routing manager 306 measures the returned packets and determines network quality parameters such as an estimated network delay (average delay) and an estimated voice data quality level (average jitter and packet loss ratio),

according to table 1 and table 2. Refer to Column 8, line 64 to Column 9, line 50. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the network-state estimation unit transmits test voice data to the second gateway apparatus via the IP network, receives test packets from the second gateway apparatus via the IP network, and determines the network-state information, including an estimated network delay and an estimated voice data quality level, based on the result of comparison of the test voice data and the test packets. One would be motivated to do so to utilize test packets to determine network qualities of various routes in the network and select the best route accordingly.

Referring to claim 14, Whitcher et al does not disclose wherein the network-state estimation unit compares a transmission time of the test voice data and a receiving time of the test packets, and calculates an estimated network delay of the IP network based on the result of the comparison of the transmission time and the receiving time.

Scott et al disclose in Figure 5 a method of testing the network quality. Upon receiving (step 510) the returned test packets from the destination gateway, the network-state estimation unit (routing manager 306) measures the returned packets and determines network average delay by subtracting the receive time from the send time, according to table 1. Refer to Column 8, line 64 to Column 9, line 50. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the network-state estimation unit compares a transmission time of the test voice data and a receiving time of the test packets, and calculates an

estimated network delay of the IP network based on the result of the comparison of the transmission time and the receiving time. One would be motivated to do so in order to utilize the test packets to determine network delay of various routes in the network and select the route with the least delay.

Referring to claim 15, Whitcher et al do not disclose wherein the network-state estimation unit determines at least one of a packet loss ratio and a packet arrival time jitter of the IP network based on the received test packets.

Scott et al disclose in Figure 5 a method of testing the network quality. The routing manager in the source gateway sends (step 508) test packets in the form of quality measurement packets to the destination gateway. The destination gateway receives (step 510) the quality measurement packets and returns the packet back to the originating gateway as soon as possible. The returned tests packet includes information about the packet that was received by the destination gateway. The routing manager measures the returned packets and determines network quality parameters such as average delay, average jitter, and packet loss ratio (table 1 and table 2). Refer to Column 8, line 64 to Column 9, line 50. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the network-state estimation unit determines at least one of a packet loss ratio and a packet arrival time jitter of the IP network based on the received test packets. One would be motivated to do so in order to determine the network conditions and select the route with the lowest the lowest average delay, average jitter and packet loss.

8. Claim 16 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 6,754,221 to Whitcher et al in view of U.S. Patent No. 6,738,351 to Qureshi et al in view of U.S. Patent No. 6,816,464 to Scott et al, and in further view of U.S. Patent No. 6,466,548 to Fitzgerald.

Whitcher et al does not disclose wherein the encoding processing unit receives the test voice data from the network-state estimation unit, and generates pulse-code-modulation encoded voice data from the received test voice data.

Fitzgerald discloses wherein the encoding processing unit receives the test voice data from the network-state estimation unit, and generates pulse-code-modulation encoded voice data from the received test voice data. Refer to Column 6, lines 22-26 and Column 7, lines 6-9. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein the encoding processing unit receives the test voice data from the network-state estimation unit, and generates pulse-code-modulation encoded voice data from the received test voice data. One would be motivated to do so since PCM is a sampling technique to digitize analog audio signals for transmission over the telephone system. Refer to Column 5, lines 45-56.

9. Claim 18 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 6,738,351 to Qureshi et al in view of U.S. Patent No. 6,754,221 to Whitcher et al, and in further view of U.S. Patent No. 7,020,263 to Bauer et al.

Referring to claim 18, Qureshi et al disclose in Figure 2 a communication apparatus comprising:

An encoding processing unit (means for performing compression in gateway 18a) encoding voice data.

A packet processing unit (means for packetizing voice data in gateway 18a) generating packets so that the packets are transmitted from the communication apparatus to a second communication apparatus (device connected to PSTN 14b). Refer to Column 1, lines 44-46 and Column 5, lines 4-7.

A quality level estimation unit (network congestion manager NCM 38) determining a quality level (number of received cells, number of cell lost, number of received octets, and number of packet loss) based on packets which are received from the second communication apparatus. As shown in Table 1 (Column 6, lines 36-64), there are several congestion indicators that are based on the packets received from the destination end. Congestion indicators include the number of received cells, number of cell lost, number of received octets, and number of packet loss. Refer to Column 5, line 59 to Column 7, line 11.

A determination unit (NCM 38) controlling the encoding of voice data by the encoding processing unit such that, when a congestion state is detected based on the quality level determined by the quality level estimation unit, a CODEC type having a compression ratio higher than a compression ratio of a CODEC type selected in a non-congestion state is selected for the encoding of voice data by encoding processing unit. When congestion decreases in the system, the voice compression ratio is decreased and when congestion increases in the system, the voice compression ratio is increased. Refer to Column 13, line 52 to Column 14, line 62.

Qureshi et al do not specifically disclose that the packet processing unit creates packets *through packetizing the encoded voice data from the encoding processing unit.*

Whitcher et al disclose in Figure 2 a gateway 18 with a packetization module 110 that packetizes packets received from the encoding processing unit (compression module 108). Refer to Column 11, lines 1-63. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include that the packet processing unit creates packets *through packetizing the encoded voice data from the encoding processing unit.* One would have been motivated to do so in order to compress the data and then convert the data into packets for transmission to the destination.

Qureshi et al also do not disclose wherein a change in the encoding controlled by the determination unit is informed from the communication apparatus to the second communication apparatus by using a packet generated by the packet processing unit.

Bauer et al disclose in Figure 3 a similar system wherein a network monitoring agent 300 determines network conditions and changes the compression ratio accordingly. Refer to Column 5, line 5 to Column 6, line 41. When a new compression mechanism is chosen, the source "...notifies the other party in the connection, referred to as the recipient, that all subsequent packets will be encoded with a new compression algorithm" (Column 5, lines 15-18). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to include wherein a change in the encoding controlled by the determination unit is informed from the communication apparatus to the second communication apparatus by using a packet generated by the

packet processing unit. One would have been motivated to do so in order for the destination to know which decompression algorithm to use to decompress the packets.

Response to Arguments

10. Applicant's arguments filed July 2, 2007 have been fully considered but they are not persuasive.

Referring to the argument that Whitcher et al disclose that the bandwidth for each customer premises equipment is predetermined and that Whitcher et al do not disclose RTCP packets (page 9, line 21 to page 11, line 9): Memory 102 stores a table (Figure 3) of customer premises information associating each customer premises equipment 14 with bandwidth and compression information. For communication with a particular customer premises equipment 14, a compression algorithm is chosen according to the available bandwidth. Refer to Column 12, line 39 to Column 14, line 4. As shown in Figure 5, the total bandwidth 302 and available bandwidth 308 determines what type of compression algorithm will be used for different subscribers 310-316. As subscribers 310-316 join the system, the bandwidth allocated for voice 306 increases and the available bandwidth 308 decreases. As the available bandwidth 308 decreases, gateway 18 selects the compression algorithm that uses less bandwidth. The network-state information (bandwidth) is based on packets received from customers. If there are more customers sending packets, the available bandwidth decreases which affects the chosen compression algorithm. Refer to Column 15, line 15 to Column 16, line 19. Therefore, the bandwidth changes. Refer also to the new rejection of claims 1 and 17.

Referring to the argument of claim 18 (page 11, lines 10-17): Refer to the new rejection of claim 18.

Conclusion

11. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Christine Ng whose telephone number is (571) 272-3124. The examiner can normally be reached on M-F; 8:00 am - 5:00 pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Huy Vu can be reached on (571) 272-3155. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

C. Ng CN
August 22, 2007

HUY D. VU
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2600